

Serial No.: 09/638,245
Amdt. dated 18 April 2007
Reply to Office Action of 31 October 2006

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REMARKS

Applicant's representative, Mr. Leslie Tyler, and the undersigned appreciate the courtesies extended to them by Examiner Lee at the interview on April 11, 2007.

Claims 7-9 and 49-120 remain in the application. Claims 7-9, 49-59 and 94-103 are withdrawn from consideration as being directed to non-elected inventions. Claims 60-93 and 104-114 are directed to the elected invention. Claims 60, 69, 73, 78, 80, 87, 107, 108, 110, 112 and 114 have been amended, and claims 115-120 have been added, in order to more clearly define applicant's invention..

Claim 110 has been rejected under 35 U.S.C. § 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention. Claims 87, 107 and 108 have been rejected under 35 U.S.C. § 102(b) as being anticipated by Todd (US Patent No 5357284). Finally, claims 60-86, 88-93, 104-106 and 109-114 have been rejected under 35 U.S.C. § 103(a) as being unpatentable over the prior art as illustrated in Fig. 1 in view of Holt et al (US Patent No. 4803727). These rejections are respectfully traversed and reconsideration is requested in view of the forgoing amendments and following remarks.

The pending elected claims 60-93 and 104-114, as well as newly added claims 115-120, are directed to systems and methods of various aspects of processing digital audio signals in accordance with the BTSC standard. In 1996 (when the application having the earliest priority date claimed in the present application was filed) digital conversion of the conditioned sum and difference signals (or "digital BTSC compliant L+R signals" and "digital BTSC compliant L-R signals") prior to modulation was not straight forward for many reasons. At the time of filing of the first application (1996), digital signal processing was still a relatively new and expensive technology, especially for consumer applications. Analog BTSC encoders prior to the present invention were quite complex, employing unusual non-linear analog devices such as log-based rms-level detectors and exponential-responding voltage-controlled amplifiers. The spectral encoder, especially, was formed by way of both feed-forward and feed-back paths around one such voltage-controlled amplifier, resulting in an even more complicated signal processing element whose functionality is difficult to simulate, much less implement in real time within the digital domain. Indeed, so great are the processing requirements of a BTSC encoder that it was

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simply not possible to implement the device on a single Motorola 56002 – the state of the art in digital signal processor (DSP) devices at the time – and the initial commercial product employing the claimed subject matter of the present application employed a second on-board digital signal processor. Digital conversion was much less widely used at that time and the available converters were not as capable or as economically feasible as is true today. It is respectfully submitted, therefore, that a DSP could not be reasonably and successfully employed to replace the type of analog functions required by an analog BTSC encoder such that it would have been obvious to one skilled in the art at the time the present invention was made.

In addition, it is submitted that the task of creating a digital equivalent for the analog adaptive signal weighting system required by the BTSC standard is less than straightforward, and in fact was quite challenging at the time the present invention was made. This difficulty also applies to creating a digital equivalent of the analog double sideband suppressed carrier amplitude modulator. These difficulties are a result of the original specification of the BTSC standard, which was developed with the assumption that the specification was carried out using continuous-time analog systems (represented in the "s-plane" of real and imaginary frequencies), and not on the basis of sampled data systems (which are represented in the "z-plane"). As is well known in the theory of sampled data systems, transforming a given analog filter (as defined in the s-plane) to a sampled equivalent (in the z-plane) inevitably involves some warping of the amplitude and phase characteristics of the filter, which warping becomes more pronounced as frequency increases closer to the sample rate. In practical systems, especially systems which are limited in computational power such as were prevalent at the time of the present invention, this warping presents a significant obstacle to proper performance of a digital BTSC system. Any variations in phase and amplitude response of a digital BTSC encoder from the ideal analog specification will cause phase, amplitude, and separation errors in the signal as demodulated and decoded in even an ideal analog receiver. This applies as well to an analog BTSC encoder paired with a digital BTSC decoder. An important aspect of the present invention was managing the inherent warping in transforming from the s-plane to the z-plane such that the digital BTSC encoder could freely be substituted for an analog BTSC encoder without requiring corresponding changes in the BTSC decoder (whether analog or digital). While various solutions to minimizing or even eliminating this inherent warping have evolved since the time the present invention was

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made, such warping was nevertheless an inherent impediment to a digital solution, and not the least bit straight forward.

With regard to the rejection of claims 87, 107 and 108 under 35 U.S.C. § 102(b) as being anticipated by Todd (US Patent No 5357284), it is submitted that Todd does not anticipate the invention as claimed. With respect to claims 87 and 107, the Examiner notes that Todd shows a BTSC encoder (205) and a form of digital modulator (212). In particular, the Examiner's attention is directed to the facts that (i) Todd provides for two separate audio signal paths, (ii) the BTSC encoder (205) and "digital audio modulator" (212) are not located in the same path, (iii) the digital audio modulator does not modulate the output of the BTSC encoder, (iv) the digital audio modulator of Todd provides a substitute signal path for the digital audio, and does not form a digital BTSC encoder, and (v) the output of Todd's QPSK modulator is fundamentally analog (specifically, it is an analog subcarrier modulated by the digital audio signal which appears at the input of digital modulator (212)). The first path includes the analog BTSC encoder (205), but not the digital audio modulator (212). In fact, a digital signal appears only in the second audio signal path shown by Todd, and the digital signal produced there is completely separate from the BTSC signal. The digital audio modulator (212) is located in this second path, but the BTSC encoder (205) is not. Furthermore, Todd's digital audio modulator (212) performs the function of QPSK modulation of the separately derived digital audio signal. In the present invention, the digital composite modulator performs amplitude modulation of a digital subcarrier by a digital BTSC signal. The output of the composite modulator disclosed in the present application is digital, not analog.

Stated in another way, Todd teaches a means of bypassing the entire analog BTSC signal path by inserting a separate analog carrier adapted to carry digital information. The analog modulation of the digital signal has nothing to do with the BTSC standard. Avoiding the analog BTSC signal path means avoiding the analog BTSC adaptive signal weighting system, including avoiding altogether the amplitude-modulated, double-sideband, suppressed carrier subcarrier for the L-R information. Todd's invention does not constitute a digital BTSC encoder in any sense, instead it constitutes a means of bypassing the analog BTSC audio transmission system altogether. It is respectfully submitted that the fact that Todd does not teach a digital version of the BTSC encoder is at least somewhat indicative of the state of the art at the time: namely that

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digital versions of a BTSC encoder were considered difficult to design, requiring substantial invention, as explained in some detail in the disclosure for the present application.

Claim 87 has been amended to recite "a system for producing a digital composite modulated BTSC signal comprising a digital BTSC encoder arranged so as to generate a digital BTSC encoded signal and a digital composite modulator." Thus, claim 87 includes a digital BTSC encoder and a digital composite modulator. New claim 115 specifically recites that the digital composite modulator is positioned in the same signal path as the BTSC encoder, and modulates so as to generate a digital composite modulated BTSC signal responsively to and as a function of the output of the BTSC encoder. Claim 107 has been amended to recite "a method of generating digital audio signals according to the BTSC standard comprising: a) accepting one or more digital audio input signals, b) performing a frequency translation of at least one digital audio signal to form at least one modified digital audio signal, and c) modifying the amplitude and phase of at least one of the digital audio signals according to the BTSC standard so as to create one or more corresponding digital audio output signals according to such standard." In contrast, as mentioned above Todd describes the "digital audio modulator" (212) as connected in a separate signal path unrelated to the analog BTSC encoder, and unrelated to processing the analog signal in accordance with the BTSC standard. The modulator of Todd thus performs a function that is fundamentally different from the modulator used in the present application.

Claim 108 is dependent on claim 107, and is patentable for the same reasons. Further, claim 108 recites "a method of generating digital audio signals according to Claim 107, wherein performing the frequency translation of the digital input signals includes performing the frequency translation by substantially 31.468 kHz." It is clear from the language that the frequency translation is occurring in the digital domain.

In summary, rather than teaching or referring to any digital methods of providing BTSC, Todd teaches methods to add a separate digital signal path along side of the conventional analog BTSC approach. Indeed, Todd makes no mention nor does he indicate that the BTSC encoder might be comprised of digital technology. Todd implicitly assumes that BTSC is an analog technology, and in column 1, line 28 refers to it in that way.

It is respectfully submitted, therefore, that Todd does not anticipate claims 87, 107 and 108.

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With regard to the rejection of claims 60-86, 88-93, 104-106, and 109-114 under 35 U.S.C. § 103(a) as being unpatentable over the prior art as illustrated in Fig. 1 in view of Holt et al (US Patent No. 4803727), it is submitted that Holt et al does not make obvious the various aspects of claims 60-86, 88-93, 104-106, and 109-114.

Regarding claims 82, 83, 86, 88, 89, 92, 104, 106, and 109-114, the Examiner states that Holt et al forms sum and difference audio signals by way of a matrix, and converts these sum and difference audio signals to a digital format in order, in part, to avoid the noise introduced by an analog band limiting filter. The Examiner posits that because the present invention has some of the characteristics of Holt system (e.g., the use of a matrix and digital conversion of audio signals), it is anticipated by Holt. It is submitted that the signal processing employed by Holt employs a very different methodology for very different reasons from that claimed in the present application. Holt, for example, creates a digital signal for transmission and reception, whereas the present invention – though using digital means to perform certain functions – ultimately results in transmitted and received signals that are analog over the intermediate medium. Furthermore, and much more importantly, Holt does not teach anything about the BTSC method, a limitation that appears in all of the elected claims currently presented.

To further illustrate the differences between Holt and the claimed subject matter, Holt converts analog signals to digital signals in order to overcome the comparatively trivial internal noise introduced by analog band-limiting filters. The transmission medium does not add significant noise because the digital signal is transmitted over a digital link. As stated in connection with FIGS. 3 and 4 of Holt:

Referring now to FIGS. 3 and 4, a second embodiment of the invention is shown in which stereo input signals are encoded so as to be transmitted over a standard 64 kbit/s digital link. The link (which is the subject of a draft CCITT [now referred to as the ITU-T] standard) utilizes 56 kbit/s for a speech channel using two-band sub-band coding plus speech channel using two-band sub-band coding plus ADPCM, multiplexed with an 8kbit/s side channel.

The present invention, on the other hand, is designed to reduce the more significant external interference associated with transmission over an impaired (e.g., noisy) analog medium (e.g., over-the-air or cable TV broadcast) or in other words, as noted in the present disclosure, so that

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"...the information can be recorded on or transmitted through a dynamically-limited, frequency dependent channel..."

To accomplish this much more difficult task, the presently disclosed system and method, (rather than simply converting the sum and difference signals to digital versions as described by Holt) utilize an adaptive digital signal weighting system (a wideband compressor and a spectral compressor which form fundamental elements of a BTSC encoder) which encodes the sum and difference signals in a sophisticated, frequency-dependent, non-linear manner prior to transmission, and decodes the encoded signal in a like but somewhat complementary manner upon reception. The system and method, therefore, by way of its particular encoding and decoding scheme, reduces the audibility of noise that is introduced by the transmission channel itself. Holt, on the other hand, simply uses digital conversion and transmits the digital signals over a digital link to reduce the self-noise generated by the analog components in its analog equivalent.

Regarding claims 60, 63, 64, 67, 68, 69, 71-73, and 76-81, the Examiner has rejected those claims under the assumption that one skilled in the art would either a) convert the digital signals to analog and then use the analog modulator shown in Fig. 1, or b) use a digital modulator to directly modulate the conditioned sum and difference signals.

As stated above, in 1996 there were many obstacles to the simple digital conversion of the conditioned sum and difference signals (or "digital BTSC compliant L+R signal" and "digital BTSC compliant L-R signal") prior to modulation. These obstacles prevent the presently claimed invention from being obvious to one skilled in the art. For example, the Examiner's attention is directed to Easley (US 6,037,993) – which post-dates the present invention – and is assigned to the present assignee – as further evidence of the foregoing. Claim 1 of Easley includes a "digital-to-analog converter for converting the digital BTSC L+R signal to an analog BTSC L+R signal and for converting the digital BTSC L-R signal to an analog BTSC L-R signal". It is submitted that if this step was not obvious at the time of Easley (filed in 1998) then it was even less so in 1996 at the time of the present invention was made.

Moreover, the present invention teaches much more than simply converting the conditioned sum and difference signals from analog to digital. Specifically, the present invention teaches conditioning a digital version of a sum and difference signal using a digital version of an adaptive signal weighting system (where the adaptive signal weighting system

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conforms to aspects of the BTSC standard). Accomplishing this in digital fashion was, at the time of the invention, anything but obvious to those who were skilled in the art.

It is submitted that digital modulators can be even more difficult to implement than digital to analog converters (DACs) due to their requirements for linearity and noise performance at significantly higher sampling rates. In 1996 digital modulators were even less available, less capable, and less likely to be considered as a practical alternative to audio-frequency DACs.

Further, while Holt performs digital conversion of analog signals and vice-versa, Holt does so specifically in order to reduce the self-noise of band-limiting analog filters and to allow subsequent downsampling and upsampling of a derived difference channel signal. In the present invention, digital conversion is utilized to overcome, among other things, the need for "complex component selection and extensive calibration required to produce acceptable analog difference channel processing sections" and to avoid "the tendency of analog components to drift, over time, away from their calibrated operation points". It is submitted that without teaching the various aspects of the claims, there is no motivation or suggestion in the reference to conclude that the invention is obvious. Therefore, Holt does not anticipate nor make obvious the use of digital conversion processes as claimed in the present invention.

Regarding claims 61, 65, 70, 74, 90, and 91, the Examiner notes that Holt fails to show any use of digital signal processing. In fact, it is unlikely that any practical design would use DSP to implement the very simple low-pass filtering described by Holt. Further, as the Examiner notes, a "big advantage of the DSP lies in the programmability of the processor, allowing parameters to easily changed." Given that Holt describes a filter with fixed characteristics (e.g. passband), such programmability is not required, and the most likely implementation would employ gates and flip-flops instead. It is respectfully submitted, therefore, that Holt does not anticipate digital signal processing recited in claims 61, 65, 70, 74, 90 and 91. On the other hand, the spectral signal weighting system that is an essential part of the BTSC transmission system constitutes a signal-dependent variable filter which lends itself much more to DSP implementations than via a fixed arrangement of gates and flip-flops.

Claims 60-93 and 104-120, directed to the elected invention, are all considered allowable over Todd and Holt et al. An early and favorable action thereon is therefore earnestly solicited.

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No further fees are believed due; however please charge any fees which may be due, or credit any overpayment, to Deposit Account Number 50-1133.

Respectfully submitted,

Date:

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